Before the
FEDERAL COMMUNICATIONS COMMISSION
WASHINGTON, D.C. 20554

In the Matter of:	)	
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A National Broadband Plan for Our Future -- Comment Sought on Defining "Broadband"	))	**GN Dockets No. 09-47, 09-51, 09-137**
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COMMENTS – NBP PUBLIC COMMENT #1

BY TELCORDIA TECHNOLOGIES

Telcordia Technologies (Telcordia) hereby submits comments to the Federal Communications Commission (FCC or "Commission") on its Public Notice requesting Comments on Defining "Broadband" in the above-captioned proceedings.[1]

In the American Recovery and Reinvestment Act of 2009[2] Congress charged the Commission with creating a national broadband plan. The Plan, due to Congress by February 17, 2010, should establish a roadmap toward achieving the goal of ensuring that all Americans reap the benefits of broadband. The Commission's Public Notice seeks "targeted" comments on the definition of "broadband" on several key aspects: (1) the general form, characteristics, and performance indications that should be included in a

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[1] Public Notice, Comment Sought on Defining "Broadband" NBP Public Notice #1, GN Dockets No. 09-47, 09-51,09-137 DA 09-1842, Released August 20, 2009.
[2] American Recovery and Reinvestment Act of 2009, Pub. L. No. 111-5, 123 Stat. 115 (2009) (Recovery Act).

definition of broadband; (2) the thresholds that should be assigned to these performance indicators today; and (3) how the definition should be reevaluated over time.

BACKGROUND

Telcordia is a software, engineering and consulting company with a vested interest in expanding the deployment of broadband. Telcordia, formerly known as Bell Communications Research (Bellcore), was created in 1984 at the time of the AT&T divestiture as a unique entity with a mission to provide common R&D as well as technology generic requirements and seamless operational capabilities across all the new service provider boundaries. We have the depth and breadth of telecommunications experience to handle the full spectrum of broadband and information network engineering and design issues. We offer the following comments on the issues raised by the Commission.

DISCUSSION

We commend the FCC for issuing this call for additional detail on the definition of broadband. This very basic question is quite nuanced, fundamentally complex, and essential to our National Broadband Plan. In our filing to the initial NOI on the National Broadband Plan from the FCC in June 2009[3] (the Initial Filing), Telcordia recommended developing a set of **Broadband Performance Indices (or Indicators) (BPIs)** similar to those successfully employed for other complex systems such as vehicle safety ratings, EPA mileage standards, Consumer Price Indices, and Energy efficiency ratings. Such

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[3] June submission on FCC Docket 09-51 Comments by Telcordia Technologies, available at: http://fjallfoss.fcc.gov/prod/ecfs/retrieve.cgi?native_or_pdf=pdf&id_document=6520219741

indices provide accurate, useful, and accessible benchmarks for consumers, businesses, and government. They are used both to track the state and to advance the progress of these underlying systems; they are also adaptable so that they can be revised and modified to reflect ongoing change and evolution.

For example, the vehicle safety rating program of the National Highway Traffic Safety Administration (NHTSA) serves both to set minimum standards for vehicles and to assess vehicle safety via a five-level star rating. For ratings purposes, vehicles are divided into classes which may be further subdivided.[4] Several different indices are calculated representing performance on crash tests (front and side) and rollover tests. Each index in turn is comprised of a number of detailed measurements and assessments; e.g., front crash test scores are based on separate injury risk curves for chest injury, leg injury, and head injury. The system is flexible and expandable (e.g., rollover testing was added in 2001). It supports simple and usable summary metrics (i.e., easy to use star ratings are calculated from detailed injury curves and Static Stability Factor computations). These summary metrics are readily understandable and widely available (see www.safercar.gov). It is also important to note that the government effort is complemented by an industry effort led by the Insurance Institute for Highway Safety (www.iihs.org) which expands the range of crash testing (rear crash testing is assessed by the IIHS and roof crush ratings are coming in 2010).

Construction of useful and usable BPIs is necessary to **transform diverse, complex, and relatively inaccessible network performance data into user-friendly ratings for convenient comparisons of broadband.**

[4] The passenger car class is divided into 8 subclasses based on type, e.g., SUV and van and, for cars, on curb weight.

In the remainder of this Comment we address the development of a framework for defining broadband and measuring its performance in the following sections:

- Summaries of our comments for each of Items 1 through 3 in the NOI (Sections I, II and II);
- Detailed material for Item 1 of the NOI (Section IV);
- Detailed material for Item 2 of the NOI (Section V);
- Conclusion;
- Three technical Appendices providing additional detail on performance standards as well as an overview of performance thresholds in current standards (Appendices A, B and C);
- Table of Acronyms and Abbreviations (Appendix D); and
- List of Technical References.

I. Summary of Comments to Item 1: Form, Characteristics, and Performance Indicators

As an overall principle, broadband definitions should be approached from the perspective of how the *user experience* is impacted by *infrastructure capabilities*, rather than the other way around. A focus on the user experience means that definitions cannot be restricted to the bit-level transport layer but must also include the packet layer of the network. The centrality of the Internet Protocol (IP) for packet transport justifies a focus on this protocol, but the concepts presented here are adaptable to alternate packet protocols. Satisfactory service delivery to end users depends on the interaction of multiple networks and systems that are often not under the control of a single entity.

BPIs relevant to the user experience are necessarily ‘end-to-end’ and this is reflected in our discussion.

We recommend broadband be defined and measured through multiple broadband performance indices (BPIs) for a set of defined user services. User services are divided into classes with the following key distinctions.

i. **Mobile** services should be distinguished from **fixed** services

ii. **Real-time** services should be separated from **non-real time** services.

iii. **Trusted Services** associated with additional, often stringent, performance parameters and requirements are further separated from standard services. Supporting critical applications in public safety, emergency services, law enforcement, health care, transportation, energy systems management, and financial transactions on a common high speed national broadband infrastructure has advantages in improving the efficiency of use of spectrum, capacity, and other network resources. These critical applications, however, require a higher level of trust in availability, security, privacy, assurance, traceability and fault-tolerance which results in additional performance parameters.

For each service type identified, a set of relevant **service quality parameters** will be defined and the BPIs will then be created via an appropriate combination of data on these parameters. In the remainder of this Comment we will focus on the service quality parameters and potential thresholds. The reason for this is two-fold. First, there has been considerable research and development within the telecommunications and information networking industry to identify, assess, and measure various service quality parameters. Second, we believe that the development of the BPIs should start first with agreement on a manageable and useful set of service quality parameters and representative services for which user experience data can be obtained and correlated to service parameters. Once these are in hand, they can be combined to form useful broadband performance indicators against which thresholds can be considered.

For standard broadband services we propose the following set of seven service quality parameters.

a. **Throughput**, both uplink and downlink and focusing on Constant Bit Rate (CBR) performance.

b. Availability including both outage and repair time.

c. Frequency and severity of **degraded service quality (DSQ)** events.

d. Packet Loss Rate (PLR).

e. Latency or delay.

f. Jitter.

g. Estimated Mean Opinion Scores (MOS), which act as a proxy for a user experience evaluation.

For *trusted services* additional service quality parameters will be required to assess: Security of information and connections; Privacy of content and transactions; High-availability; Service assurance; Identity management; and Traceability, backup and rapid restoration. These parameters derive from the specific use cases for trusted services and these use cases are discussed in more detail below.

II. Summary of Comments to Section 2: Thresholds

In the framework we are proposing, thresholds are defined for specific service quality parameters and service types. An example would be defining a threshold for acceptable packet loss rate, throughput, and jitter that together would make up the BPI for fixed real-time services. As noted earlier, service quality parameters have been the subject of very active study and research in the industry and in our detailed comments below we have provided a technical overview.

III. Summary of Comments to Item 3: Updates

Updates to the definitions, thresholds and BPIs will be needed as broadband technologies, services, devices, and applications evolve and as network performance adequacy against changing user needs also changes. The simplest update is changing a threshold (analogous, perhaps, to the EPA raising mileage standards). More complex updates involve changing the definitions or the set of underlying service quality parameters. This is analogous to changing the basket of goods in the Consumer Price Index or adding rollover ratings to crash testing.

An ongoing national effort to develop, collect, and analyze national broadband infrastructure metrics is needed. This effort will not only provide the data necessary to monitor this critical national infrastructure, but it will also support the analysis to determine when and how to upgrade thresholds, parameters and BPIs. **We recommend that the FCC make such a consistent measurement and metric activity a component of the National Broadband Plan**.

IV. Detailed Comments on Item 1: Form, Characteristics and Performance Indicators

This section provides more detailed information on item 1 of the NOI and focuses on service quality parameters for IP networks. Each of the seven proposed service quality parameters is discussed in turn, followed by a subsection on *Trusted Service* use cases and a final subsection that discusses how broadband performance indicators can be built up from the service quality parameters. IP service quality parameters have been widely studied under a variety of terminologies.[5] There is a large body of research describing and defining performance metrics for IP networks and related services by a wide variety of industry organizations [Technical References 1-38].[6] Given our focus on user-experienced performance, *the measurement point for service quality parameters is generally at the user terminal, after all error-correction, and at the presentation layer.*

We consider an initial stratification of services into two classes: **real-time services** and **non-real-time services,** which can then be further stratified to mobile, fixed, and trusted services. Typical real-time services are VoIP (voice over IP) and streaming IP video with Real-Time Protocol (RTP) or User-Datagram Protocol (UD with no re-transmission). Typical non-real time services are web browsing, file transfer, and email with Transmission Control Protocol (TCP). Other studies[7] have stratified IP services into approximately eight classes with some differences across studies. Real-time services are

[5] Some terms include: Quality of Service (QoS) metrics, Quality of Experience (QoE) metrics, Key Performance Indicators (KPIs), or Key Quality Indicators (KQIs).
[6] Relevant groups include: IETF IP Performance Metrics (IPPM), IETF RTCP XR, IETF snmpconf, ATIS IIF QoS Metrics Committee, the Video Quality Experts Group (VQEG), Video Services Forum (VSF), ITU-T SG 7, SG 9, SG12 and SG14, ITU-R SG6 WP6Q, ETSI STQ, Broadband Forum TR-126, TIA TR-30, DVB, ETSI, the TeleManagement Forum (TMF) in GB938, ATIS TMOC, ATIS PTSC SAC, and ATIS PRQC QoS, ATSC, MPEG, SCTE, 3GPP SA4 and other forums.
[7] See ITU-T Y.1541 [6], ITU-T G.1080[5], and IP or Ethernet priority mechanisms [38].

sometimes split into *one-way services* (broadcast) and *two-way services* (voice, requiring low latency).[8] The focus here is on fixed broadband access for which performance is reasonably well established but similar considerations are appropriate for both mobile as well as trusted services. The level of detail is kept to the minimum necessary to capture the major affects of broadband from a user perspective. A great many metrics exist [1]-[41], but many are quite specialized and focus on a particular application or purpose.[9] The following sub-sections describe the individual IP-layer service quality parameters as a proxy for any packet layer service. These can be recorded for individual users, service, and streams; or aggregated across users, network segments, and time in various ways [14][32].

A. Throughput[10]

Throughput is the average bit rate (number of bits successfully transmitted per unit time) on an interface or link, and is sometimes reported as the percent utilization of the overall link capacity. Throughput is calculated as an average over some length of time; when calculated over very small time lengths (below about one second) "instantaneous throughput" may be reported. More typically, speed tests calculate throughput over approximately one or two minutes. Throughput should be measured and reported in the busy hour. Real-time services should have throughput upheld during about every one second interval, otherwise packets may be lost causing degraded service. Non-real-time

[8] Non-real-time services could be split into *delay-intolerant* (web browsing) and *delay-tolerant* (background file transfer) services, however this typically is not useful.

[9] A good, somewhat extended listing of performance metrics is in ATIS-0800008 [1]; more metrics are in other references.

[10] We have elected to use throughput as opposed to goodput. In most cases the two measures will be quite similar with the main difference being treatment of overhead and management packets (included in throughput, excluded in goodput). In cases of high packet loss and retransmission, the two will differ but the packet loss rate parameter will serve to address this. A 'Goodput' BPI can be created that combines these factors.

services may have throughput upheld during intervals of up to a few minutes long without service degradation.[11] Throughput is most meaningful when combined with loss rates to assess the information delivered between end applications.

Throughput is often specified differently for upstream and downstream because of different service expectations, and thus should be tabulated for each direction. Since end-to-end performance defines the user experience, raw throughput on an individual network segment is insufficient to measure the user experience. Thus, while disaggregated segment throughput can be computed and tabulated, the actual user experience will depend on the end-to-end connection, including many factors such as network congestion, server loading and latency, and CPE. To efficiently address these factors, a modest set of canonical configurations can be used. ITU G.1050 assessed a number of such configurations in the form of User <–> Access <–> Core <–> Access <–> User [33].

B. Availability

Availability or uptime is the percentage of time that the broadband service is available [5]. Downtime includes unexpected outages as well as scheduled downtime for maintenance.[12]

[11] IP throughput in connection agreements is commonly defined in terms of Committed Information Rate (CIR) and Excess Information Rate (EIR). The CIR is guaranteed while the EIR is provided if there is adequate bandwidth. CIR is the average bandwidth for a virtual circuit guaranteed to work under normal conditions. At any given time, the bandwidth should not fall below the CIR. The EIR is an allowance for short bursts of bandwidth above the CIR. The CIR plus EIR is less than or equal to the speed of the access line into the network. The EIR is sometimes also called the Peak Information Rate (PIR). The CIR is sometimes called the Committed Data Rate (CDR). CIR and EIR can be specified to be measured and achieved during a certain time period.

[12] Availability has sometimes been estimated by aggregating statistics for reliability and mean-time-to-repair (MTTR) of equipment and links comprising a network [14][32]; however, in a system as diverse as the Internet this is impractical.

C. Frequency and Severity of Degraded Service Quality (DSQ) Events

TM Forum GB938 Version 2.0 [28] defines a Degraded Service Quality (DSQ) event as “a noticeable impairment of the audio quality, video quality, or service response time.” GB938 lists a number of different DSQ event types and defines metrics based on these, such as the percent of session time with service quality lower than a threshold. This definition is broadened somewhat here to include network degradation as well as service degradation. A DSQ event could simply be an errored second (ES) or severely errored second (SES). For non-real time services a DSQ event could be defined as an occurrence of the usable throughput dropping below half of its nominal value. For real-time services a DSQ could be defined as a level of network impairment sufficient to cause a noticeable degradation to service quality.

Frequency counts of DSQ events capture an important component of service quality. Note that DSQ events are not included in availability measures which only encompass complete service outages.

D. Packet Loss Rate (PLR)

The Packet Loss Rate (PLR) is the number of lost and discarded packets divided by the number of transmitted packets [24]. The PLR scales with the bit rate since it is per transmitted packet and not per unit time. Packets are typically lost because they experience lower-layer errors or because they were discarded at a congested intermediate switch or router queue. We recommend that packet loss be measured after error correction and retransmission to focus on user-experienced performance.[13] Note that for

[13] Lost packets are usually identified by missing sequence numbers so that assessing PLR requires protocols with sequence numbers and access to those.

non-real time services using TCP, the PLR affects the throughput because re-transmissions slow down TCP. For these services, only throughput needs to be assessed. Packet loss is often bursty; e.g., losing many packets in a short burst of time because of brief times of congestion.[14] Although the same PLR with different burst behavior can have a much different effect on a given application, simply assessing the PLR is generally sufficient.

E. Latency

Latency or delay is important particularly for two-way real-time applications. For example, end-to-end delay for VoIP should be bounded to enable conversation. One-way delay can be difficult to measure. Round-trip delay (measured with an IP ping for example) is much easier to measure but generally requires an active test. For many applications, session initiation delays or server delays dominate one-way network delay.

F. Jitter

Jitter, also called delay variation can be computed in a variety of different ways and in relation to longer or shorter term average packet inter-arrival times. The definition of jitter in RFC 3550 is recommended here [22]. Jitter is an estimate of the statistical variance of the data packet inter-arrival time. The inter-arrival jitter is defined to be the mean deviation (smoothed absolute value) of the difference in packet spacing at the receiver compared to the sender for a pair of packets. Excessive jitter can cause buffer overflow or under-run which degrades the performance of real-time applications. Jitter

[14] Burst loss can be characterized by time-series behavior, often using a Gilbert-Elliot model, with a low-loss "gap" state and a high-loss "burst" state. Packet loss burst metrics include [23][24]: Burst length: average time duration of burst error events; Gap length: average time between bursts; Burst rate: PLR during bursts; Gap rate: PLR during gaps; Error-causing events: frequency of rapidly occurring or long bursts, defined by crossing thresholds, etc

can be measured from Real-Time Protocol (RTP) or MPEG[15] transport streams (TS) time stamps. Jitter impacts both two-way and one-way real-time services; however it has minimal impact on non-real-time services.[16]

G. Estimated Mean Opinion Scores

The Mean Opinion Score (MOS) scale is used to measure user-perceived audio, video, or multimedia quality [24]. As seen in Table 1, MOS ranges from 5 (Excellent) to 1 (Bad).[17] MOS can be measured by a number of human subjects, however this is time-consuming and often impractical, so algorithms have been developed that estimate MOS. For VoIP, it is recommended to use the E-model R-factor [11], or the PESQ algorithm [12][13] to estimate audio quality MOS. For IP video, there are a number of candidate algorithms that estimate MOS with reasonable accuracy[18], many of which have been tested by VQEG [26][27]. Because the MOS scores and other similar concepts are based on user experiences, they are important references against which proposed BPIs can be scaled.

Table 1. ITU-R BT.500 Quality scale (MOS)

Quality (MOS)	
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

[15] MPEG stands for Motion Picture Experts Group and refers to a set of standards for video encoding.
[16] There is often a trade-off between jitter and latency in that jitter can be reduced by using a large receive buffer in the CPE at the price of increased latency.
[17] There are actually several MOS ratings with MOS-V estimated user-perceived picture quality [8]. MOS-A estimated the user-perceived audio quality [11][12] and MOS-AV [24] estimates Audio/Video or multimedia quality.
[18] That is, btter than using a simple Peak Signal to Noise Ratio (PSNR) [7][10].

MOS may be objectively estimated with full-reference, reduced-reference, or no-reference methods. Full-reference MOS methods have access to both the original signal and its encoded counterpart. Reduced-reference methods have access to the encoded signal, but have limited knowledge of the original signal. No-reference methods have access to the received encoded signal only. No-reference methods are usually deployed for real-time, in-service quality monitoring. Full reference methods may be useful to measure the quality of service prior to transmission (e.g. at the head-end).

H. Use Cases for Trusted Services

Trusted Services necessarily encompass a wide range of scenarios and applications characterized by a particular need for one or more characteristics that are both not typical of standard services and also often challenging to provide. Support for Trusted Services involves reducing the prevalence of -- and assuring resilience to -- cyber-attacks, encryption of content against unintended or deliberate eavesdropping, and identity theft. Support for Trusted services also involves high-availability for both single and multi-network domain configurations, QoS management for high priority emergency service communications, security of personal information, validation of proper security protocol implementation, and verification and management of identify. As a means of identifying the key service quality parameters specific to various Trusted Services, we recommend the Commission develop a series of 'Trusted Use Cases' to illuminate the requirements and highlight necessary efforts to implement solutions for these services. The Trusted Use Cases will also inform any necessary rule-making to ensure that Trusted Services are not inadvertently excluded from participation in the national broadband infrastructure. A brief and partial list of such use cases is provided below to guide further development.

- Health Care
    - Tele-presence for maintenance of independent living
    - Medical consultation and prescription provision and verification over realtime or non-realtime services
    - Medical image exchange with traceability, backup, and identity management
- Law Enforcement
    - Incarceration avoidance with offender remote monitoring
    - Secure and private cloud computing for identity database searching
- Energy Management
    - Power grid load management
    - Remote management of energy consuming elements (smart infrastructure)
- Emergency Services
    - Maps of street view with secure access plans and maps for emergency first responders.
    - Maintenance and re-establishment of communications infrastructure in event of natural disaster.
    - On-site cross-jurisdictional communications and command-post operation
- Public Safety and National Security
    - Broadcast public safety communications to a defined geographic area with ability to target and reach the full area
    - Secure virtual border patrol inspection and remote sensing telemetry
    - Incoming and outgoing cargo inspection with sensor telemetry

I. Creating BPIs from Service Quality Parameters

Appropriately combining previously identified service quality parameters of throughput, availability, DSQ events, jitter, latency, and packet loss, we can construct a single BPI for a specific type of service. This BPI could then be compared against a user experience score to understand how a certain BPI rating would correspond to a given user experience. As in other domains, easy to use BPIs may catch hold in the industry so that, for instance, equipment might be advertised and sold with specified BPI ratings for a given broadband network. In this sense, well chosen BPIs could stimulate greater use of broadband services by removing ambiguity and reducing market confusion, important benefits given the economic leverage provided by broadband technology. [19]

A worked example of constructing a BPI for IP video quality is provided below. Methods for characterizing the constituent service quality parameters of the BPI should be defined and, if possible, automated techniques for collecting that information devised. This would permit users to understand how their individual network connection can be expected to perform under a set of standard conditions. Initially, there will be a need to understand what a given video BPI means for user experience under different operating environments (e.g. mobile, fixed, large display, high speed video interface, etc.). More

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[19] There have been numerous studies linking broadband penetration to economic impact. Among them are a study by the Brookings institute; Robert Crandall, William Lehr and Robert Litan, “The Effects of Broadband Deployment on Output and Employment: A Cross-sectional Analysis of U.S. Data”, Issues in Economic Policy, Number 6, July 2007. http://www.brookings.edu/~/media/Files/rc/papers/2007/06labor_crandall/06labor_crandall.pdf, from Connected Nation:, “The economic impact of stimulating broadband nationally, A report from Connected nation,” Feb. 21, 2008, http://www.nga.org/Files/pdf/0812BROADBANDCONNECTED.PDF, and a study s sponsored by the U. S. Economic Development Administration of the Dept. Of Commerce, “Measuring the Economic Impact of Broadband Deployment,” Final Report National Technical Assistance, Training, Research, and Evaluation Project #99-07-13829 Sharon E. Gillett (sharoneg@mit.edu), Principal Investigator, Dr. William H. Lehr, Carlos A. Osorio, Massachusetts Institute of Technology, Feb. 28, 2006, http://www.eda.gov/PDF/MITCMUBBImpactReport.pdf

specialized performance metrics for Trusted Services such as telemedicine and public safety will need further definition. This exercise is not without precedent. For example ITU-T G.1050 [33] (and [32]) describes a method of categorizing the likelihood of occurrence of different network impairments, and includes a process to determine overall network performance based on the implications for end-user applications.

Generally BPIs are created by aggregating the individual service quality parameters described in this Comment.[20] In Table 2, several concepts for BPIs are created by combining the lower-level individual service quality parameters with an "X" in the table to create the aggregated BPI. In this table, we have omitted the MOS score, treating it ideally as a dependent result of the other performance indicators, even though this is not necessarily the case if special network provisions are made just to boost MOS scores. Just as one may create additional BPIs to address specific service types, so too can one incorporate additional service quality parameters. Study of user experience measures against objective network performance data is an active area of research and further study may uncover additional or more appropriate data that are reflected in user experience. These can be incorporated over time to fine tune desired BPIs and to define specific relations among parameters to define the index. An example of this exercise for a 'Trouble Level' indicator is described as a quantitative example.

[20] In the jargon of the industry, this is similar to aggregating key performance indicators (KPIs) into key quality indicators (KQIs).

Table 2. Examples of aggregation of service quality parameters into higher-level Broadband Performance Indices (BPI).

Service Quality Parameter / BPI	Throughput	Availability	# DSQ events	PLR	Jitter	Latency	Security, Privacy Indicators
Overall – Mean Performance Guarantee	X	X	X	X	X	X	X
Service Quality Index			X	X	X	X	
"Goodput" Index (non-real-time service index)	X			X			
Uptime Index		X	X				
Video Performance Index (VPI)	X			X	X	X	
Trusted Service Index (TSI)		X	X				X

DSQ- Degraded Service Quality
PLR - Packet Loss Rate

As a second and final example, we describe a method for assessing the impact of errors on the user-perceived quality using a two-stage process. This process accounts both for the severity and the frequency of error or degraded service quality (DSQ) events. The specific numbers in the tables here are for illustration only. Severities are defined on a scale from 1 (worst) to 5 (no trouble) to be consistent with an MOS score, but are model values and do not reflect actual MOS scores.

First, we define the severity level of each error event depending on its duration and extent, as shown in Table 3:

Table 3. Severity level of a single error. The example is for a video or audio service, and some percentage of the picture or audio is impacted by an error.

Duration of error	Extent of picture & audio impacted	Severity of error
< 1 second	< 10%	4
< 1 second	between 10% and 35%	3
< 1 second	> 35%	2
between 1 and 5 seconds	< 10%	3
between 1 and 5 seconds	between 10% and 35%	2
between 1 and 5 seconds	> 35%	1
> 5 seconds	< 10%	2
> 5 seconds	between 10% and 35%	1
> 5 seconds	> 35%	1

Then, we aggregate all the occurrences of error events accounting for their severity levels and frequency of occurrence as shown in Table 4:

Table 4. Aggregating multiple errors into an overall Trouble level.

Number of perceivable errors per hour	Severity of error [Table 3]	Trouble level
< 0.25	Any	No trouble
between 0.25 and 1	3 to 4	No trouble
between 0.25 and 1	1 to 2	Minor
between 1 and 2	4	No trouble
between 1 and 2	1 to 3	Minor
between 3 and 5	4	Minor
between 3 and 5	2 to 3	Major
between 3 and 5	1	Critical
> 5	4	Major
> 5	1 to 3	Critical

Note that the qualitative ‘No Trouble, Minor, Major, and Critical’ scales could be converted to numbers or ‘star’ ratings if desired. Levels can be initially established based on subjective or objective tests, call center data, survey data, or engineering requirements. Over time, once a user experience level has been set, it can be adjusted based on a balance of alarms and customer complaints.

V. Thresholds

The setting of thresholds is fundamentally different for non-real-time services versus real-time services. The basis of thresholds should be user experience, and several service parameters will factor into a given user experience. In the first three subsections below we very briefly discuss setting thresholds for three service quality parameters most relevant for non-real-time services; namely, Throughput, Availability and Degraded Service Quality. In the next two subsections we consider in greater detail thresholds for Packet Loss Rate (PLR), Latency, Jitter, and Mean Opinion Scores (MOS) by focusing on two specific real-time services: VoIP and IP Video. We conclude this section with a discussion of service quality parameters and thresholds for wireless networks. Additional details on several of the technical topics in this Section are provided in the Appendices, with some current standard levels abstracted in Appendix A, providing a view of both the specificity and the variability that is seen. Appendix B provides material on VOIP QoS and QoE measures and Appendix C gives additional detail on IP Video metrics.

A. Throughput

Throughput or bit rate is often specified for broadband access lines; however the advertised throughput is often not achieved in practice, particularly for wireless services. ITU-T G.1010 [4] describes different non real time services; broadly classified as data services or background services, and lists some performance targets for these: One-way delay, jitter, and information loss. One-way delay targets are 2 to 4 seconds for web browsing and 15 to 60 seconds for bulk data transfer. Although this is somewhat circular, these could be used to calculate throughput targets since the throughput often determines how fast a web page can load or a file can transfer. However, data transfer size is highly variable, and throughput targets are elusive.

Throughput may have to change significantly depending on factors including

- Richness of web page content
- Size of bulk data transfer
- Number of data consuming elements in the CPE network
- Type of application (Trusted Services may require higher bulk transfer for uncompressed images, for example.

All of these factors can significantly affect demand on an access link, and can be expected over time to put upward pressure on throughput. Finally, upload speed is important for some applications such as file sharing, but not important for others.

Geographic characteristics also factor into the ability to deliver a given throughput, and thus may need to be taken into consideration when evaluating throughput thresholds. It is often more challenging technologically and economically to deliver high throughput over sparse rural areas than it is in high density urban environments. In a country as diverse as the United States, it is appropriate to be able to geographically resolve and measure both throughput and related BPIs so that one can assure that all areas continue to improve in throughput over time.

Finally, we must bear in mind that the United States competes in a global economy in which broadband infrastructure is a significant factor. Any definition of broadband that ignores this fact will quickly become obsolete. Measurement of BPI and broadband definitions and thresholds should therefore also include an element of international comparison to ensure that the United States maintains competitive strength in this important dimension of economic effectiveness.

B. Availability

A reasonable threshold for fixed Internet access would be roughly 99% availability and for mobile access, 95% availability. Availability thresholds should be much higher for Trusted Services and the ability to establish priority for such services and assure their delivery will be critical.[21]

C. Frequency and Severity of Degraded Service Quality (DSQ) events

Degraded Service Quality (DSQ) is frequently experienced by Internet users, and there are many ways to precisely define DSQ events [28]. For non-real-time services, a simple specification such as noticeable DSQ occurs less than 10% of the time may suffice. Achieving a smaller rate of DSQ events is generally necessary for real-time services.

It is common to specify the number of perceptible error events per hour, for example Broadband Forum TR-126 [29] specifies no more than one loss event each one to four hours. Thresholds on percent of DSQ time could be specified based on the Packet loss rates (PLR) in Table B **-- 13** and Table C **-- 14** in Appendix A. For example, ITU-T J.241 [4] recommends percent DSQ times from 0 to 0.2% percent for digital video.

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[21] Similarly, it will be necessary to consider *all* scenarios in which a Trusted Service needs to operate. There may be a tendency to restrict thinking to scenarios for which existing systems are designed to operate and exclude others. For example, inter-domain emergency service communication is often simply not possible, witnessed by situations in which fire-fighters and police officials cannot communicate with one another over dedicated networks. In this case, Availability in the formal sense is zero.

DSQ can expand on the number of errors per hour by also accounting for the severity of each error event. Different impairments may occur at differing severities, over different lengths of time intervals, and with different rates of occurrence. Severity levels for impairments have also been developed. ITU-R BT.500 [8] defines subjective tests for impairment scales, as shown in **Error! Reference source not found.**Table 2.

Table 5. ITU-R BT.500 and Impairment scale

Impairment scale	
1	Very annoying
2	Annoying
3	Slightly annoying
4	Perceptible, but not annoying
5	Imperceptible

A stratification of alarm levels typically used by the industry is presented in **Error! Reference source not found.**Table 3. These types of alarm classifications are common for monitoring physical layer links, interfaces, and equipment. **Error! Reference source not found.**Table 2, **Error! Reference source not found.**Table 3, and **Error! Reference source not found.**Table 4 show numerical scales from 1 (worst) to 5 (best). These roughly correspond between the tables, with **Error! Reference source not found.**Table 2 and **Error! Reference source not found.**Table 3 showing a somewhat generic ranking and with **Error! Reference source not found.**Table 4 originally developed for MPEG transport streams [35][36].

Table 6. Alarm Levels.

Critical	1	Worst
Severe	2	
Major	3	
Minor	4	
No trouble	5	Best

Table 7. Error Characterization.[22]

Program Off Air (POA)	1	Worst
Component Missing (CM)	2	
Quality Of Service (QOS)	3	
Technically Non-Conformant (TNC)	4	
No Error	5	Best

D. Packet Loss Rate (PLR)

Packet loss rate (PLR) is measured after all error correction and re-transmission so that it is aligned with the experience of the user or application. PLR is critical for real-time applications. Existing standards specify differing packet loss rates for IP video. ITU-T Y.1541 [6] recommends a PLR threshold of 4×10^{-7} for access distribution at 3 Mbps assuming that 10 performance hits per day are tolerated. Broadband Forum TR-126 [29] presents a range of PLR thresholds; for H.264/ MPEG4 compression PLR thresholds are specified from 6×10^{-6} (SDTV) to 1.2×10^{-6} (HDTV). For digital television, ITU-T J.241 [4] recommends that PLR $\leq 10^{-5}$ for excellent service quality (ESQ) and PLR $< 2 \times 10^{-4}$ for intermediate service quality (ISQ). Overall, existing standards recommend PLR thresholds from 2×10^{-4} to 4×10^{-7} for high-quality IP video. Recent work and published reports have shown that many loss events are not noticeable by the user, so the

[22] **Program off Air (POA):** A main service (virtual channel) is flawed to the point that that service is effectively off air for conformant/reasonable receiver designs. Receivers will not be able to tune and decode anything within the transport.

Component missing (CM): One of the program components that is signaled by previous tabular data (in MPEG TS PSIP or PSI) as present is either not present or cannot be found and decoded.

Quality of Service (QOS): Parameters are out of specification by such a margin that a significant fraction of the receivers can be expected to produce flawed outputs. In many cases, the broadcast is viewable, but may exhibit some form of degradation to the viewer.

Technically Non-Conformant (TNC): Violates the letter of the standard, but in practice will have little effect on the viewing experience. Errors of this type should be corrected, but do not have the urgency of higher severity errors.

standardized numbers are slightly rounded up here. For IP video, packet loss rate (PLR), after all error correction is recommended to be no higher than 10^{-3} to 10^{-6}. This is a somewhat wide range but the impact of PLR varies widely with service level or screen size, and the error handling capabilities of the decoder.

VoIP can convey intelligible speech at a PLR of several percent, depending on the type of codec. However, this is the barest minimum, and so a PLR no higher than 1% is recommended here for VoIP.

E. Latency

Latency is defined as the one-way delay from source to sink, and includes all the delay up though the application. The delay of the encoder is included. While some reasonable latency target could be specified for most applications, latency is generally only critical for two-way real-time services such as VoIP, video teleconferencing, or remote control (teleoperation). Specified targets for VoIP latency range from 100 to 200 milliseconds. A performance threshold of 150 millisecond latency is recommended for two-way real-time services here. Additional details on latency for IP Video are provided in Appendix C.

F. Jitter

Jitter, or delay variation, is critical for real-time applications. Broadband Forum TR-126 [29] recommends a jitter threshold of 50 milliseconds, but current IP video set-top boxes can reportedly tolerate up to 50 to 150 milliseconds jitter. CPE for services on a groomed network, such as IPTV, may only tolerate low jitter levels. CPE for services on the open Internet, such as Internet-sourced video, may be able to tolerate high jitter levels. ITU-T G.1050 [33] recommends a jitter threshold of 50 milliseconds for well managed networks and 150 milliseconds for partially-managed networks. VoIP and video conferencing are not tolerant of delay and a jitter threshold of 50 milliseconds is often specified for VoIP. An average jitter threshold of 100 milliseconds is recommended here for IP video. An average jitter threshold of 50 milliseconds is recommended here for VoIP.

G. Mean Opinion Scores (MOS)

Mean Opinion Score (MOS) is a subjective rating of content quality. MOS only generally applies to video, audio or multimedia. For VoIP, a MOS of 4 is considered PSTN quality, 3 is reasonably acceptable and 2 or less is not tolerable. In the network, MOS is generally estimated with no-reference methods using measurable parameters such as encoder settings and packet loss. These estimates are approximate, and methods of measuring compliance with MOS specifications must account for this. A threshold of a minimum MOS of 3.6 is recommended here, which originates from VoIP requirements, and is between “fair” and “good” but slightly closer to “good.”

H. Thresholds for Broadband Wireless Service Quality Parameters

Previous sections have focused on fixed broadband access. While the same broadband performance indicators and service quality parameters used for fixed access can also be used for mobile networks, thresholds should be relaxed for mobile services. As with fixed service, a distinction between uplink and downlink is particularly noted.

The following QoS metrics are included in 3GPP specifications: Packet-switched Streaming Service (PSS) in 3GPP TS 26.234 [40] and Multimedia Broadcast/Multicast Service (MBMS) in 3GPP TS 26.346 [41]:

- Corruption duration

- Rebuffering duration
- Initial buffering duration
- Successive loss of RTP packets
- Frame rate deviation
- Jitter duration (time that jitter is above 100 milliseconds)

Throughput: Reported data rates for broadband wireless are often inaccurate; they are often related to a gross line data rate that is difficult to achieve except under ideal conditions, and may not reflect the average rates available to a single user in a multi-user environment.

There are four main types of in-service testing for cellular mobile networks:

Air Interface. Drive testing or high numbers of dropped calls or blocked calls at a cell site or base station can indicate that re-configuration or new resources are needed.

Device management. Devices can have various types of faults or mis-configurations, and it is becoming more common to remotely manage devices [39].

Services testing. Unlike broadband Internet, services on cell phones are often under the auspices of the network provider. End-to-end service testing is often needed to resolve customer complaints even if they originate at external websites.

Most of the new broadband wireless technologies (e.g. WiMAX, LTE) are all-IP networks. To that extent, they are similar to fixed broadband networks. However, there are still challenges in meeting the connection-related performance objectives for session-oriented services, such as VoIP or video telephony. While meeting the broadband IP transport requirements (after a connection is established between end users or between end user and an application for session-oriented services), all the connection-related performance thresholds, such as call setup time, call drop rate, call blocking rate, handover delay, need to be kept.

Wireless backhaul, refers to the techniques used to connect cellular radio base stations to fixed line infrastructure. Formerly dominated by leased line services, these are largely migrating to Ethernet / IP backhaul.

Once established, these network connections are similar to monitoring business services, except for a higher emphasis on delay and synchronization. As with many fixed services, a broadband wireless service is supported by multiple access networks, backhaul networks, and core/backbone networks, all of which need to be broadband. In some cases, these networks are operated by different operators. As mentioned in our summary an appropriate set of BPIs and canonical use cases will be needed to gain insight into individual segments of the end-to-end metrics.

Additional performance factors for mobile and wireless physical layer: Wireless performance measurements have additional conditions to consider, such as the degree of mobility (e.g. pedestrian, car, high-speed train), the radio environment (e.g. city, suburban, rural). Further, since radio bandwidth is scarce compared to fixed networks in the access portion, wireless broadband has a particular need to schedule services based on QoS Classes (e.g. Conversational class, Streaming class, Interactive class, and Background class in 3GPP). Since broadband networks still need to support a wide spectrum of services with different throughput, delay, and jitter (only broader bandwidth compared to narrowband networks), we may need to consider selecting a "representative" set of services (e.g. VoIP and video telephony in the Conversation class, video and audio streaming in the Streaming class, ….) and their associated performance measurements (e.g., buffering delay in the streaming case) to provide an overall view of a broadband network administrated by a service provider.

CONCLUSION

Telcordia recommends that the FCC incorporate the following key concepts into the definition of broadband:

- Broadband definitions should be approached from the perspective of how the *user experience* is impacted by *infrastructure capabilities* and not the other way around. A focus on the user experience means that definitions cannot be restricted to the bit-level transport layer but must also include layers at and above the network packet layer. It also means that defining different performance thresholds for different service classes and canonical network configurations is critical. Lastly, it requires that the definitions and thresholds be dynamic and flexible so that they can evolve with the evolution in broadband infrastructure, devices, services and applications.
- We propose an initial simplified framework for defining service classes based on distinguishing *non-real-time and real-time* services, *fixed and mobile* service, and *standard and trusted* services. While more complex divisions have been developed, we believe that these three factors are sufficiently comprehensive to yield a practically useful understanding.
- A tremendous amount of research and study has been done in the telecommunications and information networking communities on broadband service quality parameters. We have endeavored to provide, particularly in the appendices, pointers to this work. Leveraging this work, much of which has taken place in standards and other industry

bodies, is essential to meet the tight time frames to develop the National Broadband Plan. We propose a set of seven service quality parameters, all of which have been addressed in the industry, as an appropriate initial set for standard services: *throughput, availability, frequency and severity degraded service quality events, packet loss rate, jitter, latency,* and *estimated mean opinion score (MOSs*. We believe these parameters provide adequate starting coverage. Clearly additional parameters are available, and could be added if shown to significantly impact the user experience that grounds our view of broadband definitions. For trusted services, additional parameters related to security, privacy, assurance and restoration will be needed and we suggest these parameters be developed based on use cases.

- Easy-to-use broadband performance indices (BPIs) would be derived through combinations of service quality parameters. Similar approaches are very effectively used in other domains, such as vehicle safety and mileage ratings. The objectives in developing these BPIs are: to transform technically complex performance data into use-friendly ratings; to produce useful quantities for benchmarking the status of broadband in different regions and communities; to use these indicators to drive the expansion of broadband capability across the country by setting targets; and to provide accessible and usable information to consumers and organizations, both public and private, for understanding and comparing broadband infrastructure options.
- It is clear that an on-going nation-wide effort on broadband data collection, measurement, and analysis is necessary. While a part of this need will be addressed through current state-level broadband mapping projects, critical work remains to

produce accurate and useful data to monitor this critical national infrastructure and to successfully implement the National Broadband Plan and track its progress. We recommend that the FCC include this ongoing effort as part of the National Plan through the establishment of a functional *Broadband Information Administration*. Among the activities of this administration will be: development of common terminology and meta-data to allow integration of diverse broadband information; operation of a publically available information repository for the use of citizens and organization; evaluation of broadband impacts, and the ongoing responsibility for producing and updating the parameters, thresholds, reference service cases, user experience data, and BPIs used to track and manage our national broadband infrastructure.

We hope that the comments and technical details we have provided will be of value to the critical task of the Commission in developing the National Broadband Plan.

Respectfully submitted,

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August 31, 2009

APPENDIX A: OVERVIEW OF CURRENT STANDARDIZED THRESHOLDS FOR BROADBAND PERFORMANCE

To provide an example of both the level of **specificity** as well as the **variability** evident in the current standardization efforts of IP based services, we provide extracted information from several related standards in this Appendix.

A. ITU-T Y.1541, Network performance objectives for IP-based services

ITU-T Y.1541 [6] is a recent standard defining performance and classes of network Quality of Service (QoS) for IP-based services. Methods of computing performance metrics and impairment accumulation are defined. Most of ITU-T Y.1541 defines performance of general Internet service classes 0 to 5. However, Appendix VIII discusses the effects of IP network performance on digital television transmission QoS, including the loss ratio recommendations shown in Table A -- VIII.1/Y.1541. ITU-T Y.1541 also recommends FEC parameters that can help achieve these targets.

Table A -- VIII.1/Y.1541 – Digital television loss/error ratio recommendations

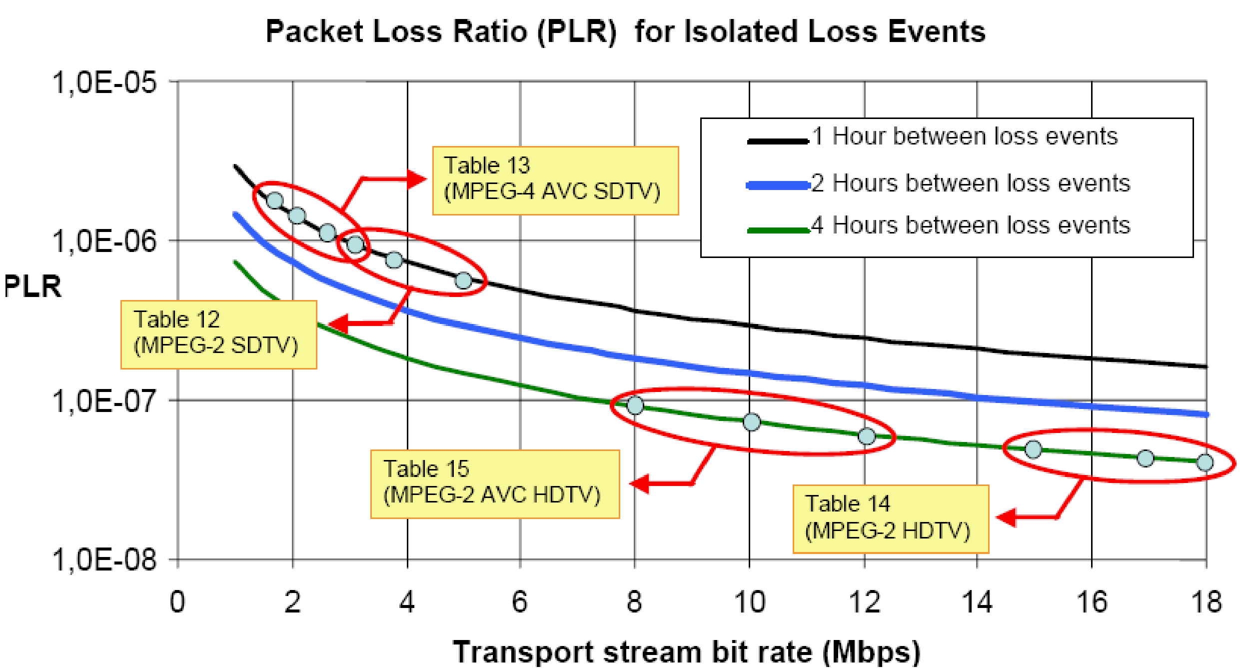
Profile (Typical bit rate)	One performance hit per 10 days	One performance hit per day	10 performance hits per day
Contribution (270 Mbit/s)	4×10^{-11}	4×10^{-10}	4×10^{-9}
Primary Distrib. (40 Mbit/s)	3×10^{-10}	3×10^{-9}	3×10^{-8}
Access Distrib. (3 Mbit/s)	4×10^{-9}	4×10^{-8}	4×10^{-7}

B. Broadband Forum (BBF) TR-126, Triple-play Services Quality of Experience (QoE) Requirements

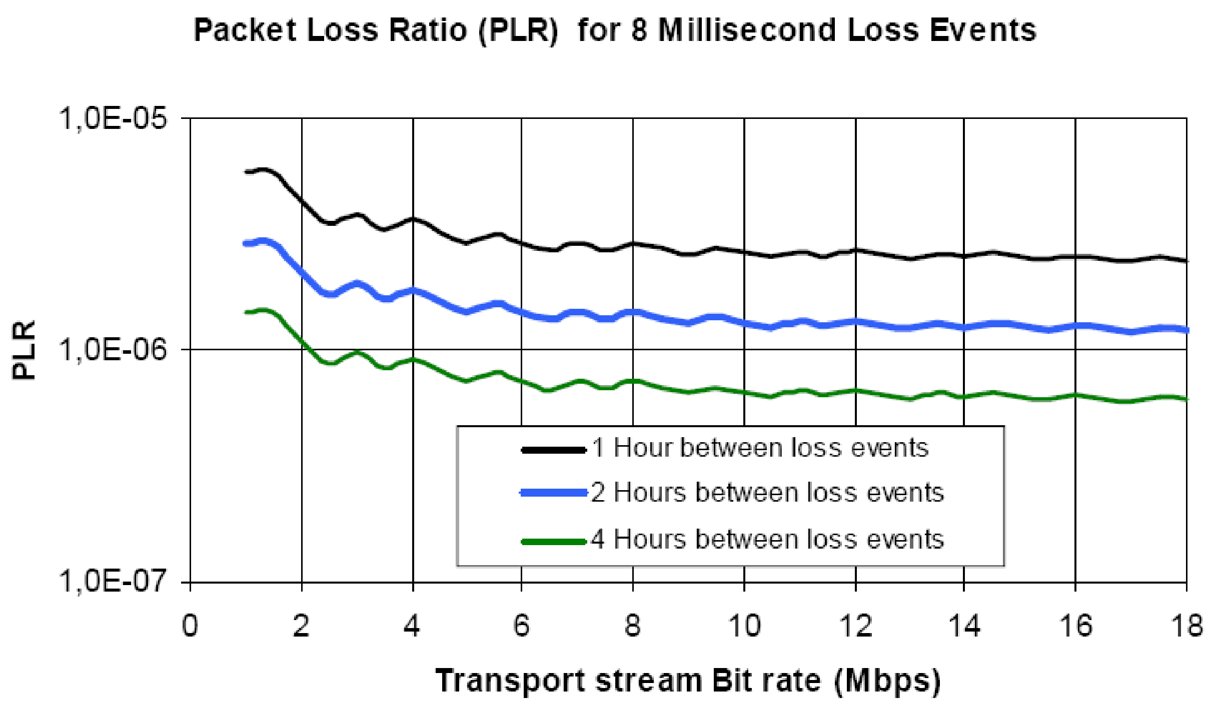
Broadband Forum TR-126 [29] discusses in detail the QoE of triple-play services, and presents many performance thresholds called QoE objectives. TR-126 discusses many services, including IP video.

For IP video, one-way latency is recommended to be at most 200 milliseconds, with jitter below 50 milliseconds. Packet loss thresholds are different for MPEG-2 and MPEG-4 compression, and for HDTV and SDTV.

IP video quality requirements are often stated in TR-126 as a "maximum" of 1 error event per time period – for example DVB's requirement of a maximum of 1 visible artifact in an hour. This is different from the "average" time between events used in the plots presented here. For example, if losses are independent and random then an average time between losses of 4 hours implies that the "maximum one loss in a hour" target will be broken roughly once per day (i.e. two errors are seen within the same hour roughly once per day). Equally, an average time between loss of half an hour means that there will very often be more than one loss in any given half hour period. Recent information has shown that subscribers may only notice a few loss events, and so the loss requirements shown here could be loosened, possibly by up to an order of magnitude [42]. The next two Figures address PLR.



BBF TR-126 Figure 12: PLR, after all error correction, required to meet average time between loss events of 1, 2, and 4 hours assuming isolated lost packets. 1 hour between loss events may apply to standard-definition broadcast-quality IP video, while 4 hours between loss events may apply to high-definition quality IP video.



BBF TR-126 Figure 13: PLR required to meet average time between loss events of 1, 2, and 4 hours assuming each event is an uncorrectable DSL error that loses 8 milliseconds of contiguous data. 1 hour between loss events may apply to standard-definition broadcast-quality IP video, while 4 hours between loss events may apply to high-definition quality IP video.

C. ITU-T J.241 Appendix A

ITU-T J.241 [4] presents some performance targets for different classes of digital video services delivered over broadband IP networks

Table A -- 8. Example informative classification used for digital television services, from ITU-T J.241 Appendix A. [4]

Packet loss rate (PLR)	QoS
$PLR \le 10^{-5}$	excellent service quality (ESQ)
$10^{-5} < PLR \le 2*10^{-4}$	intermediate service quality (ISQ)
$2*10^{-4} < PLR < PLR\_out = 0.01$	poor service quality (PSQ)
PLR_out = 0.01 < PLR	IP end-to-end service not available.

The percent of time with at least intermediate service quality or at least poor service quality is 0% to 0.2%, depending on the service class.

D. TM Forum GB938, Application Note to SLA Management Handbook Video over IP / Wireline & Wireless

TM Forum GB938 Version 2.0 [28] defines Key Quality Indicators (KQIs), which are related to QoE metrics. A Degraded Service Quality (DSQ) event is defined as "a noticeable impairment of the audio quality, video quality, or service response time." GB938 lists a number of different DSQ event types and defines KQIs based on these, such as the percent of session time with audio or video quality < X (on MOS scale).

E. ITU-T G.1080

ITU-T G.1080 [5] presents Quality of Experience Requirements for IP video. Different QoE dimensions are presented and discussed; these include objective QoS measures (service factors, transport factors, and application factors), as well as subjective human components (emotions, service level, billing, experience, etc.). Picture quality is discussed. An example set of requirements is presented for the minimum acceptable bit rates of compressed video and audio. Tables are given for MPEG-2 and H.264

compression; these are stratified for HDTV and STDV, and also for broadcast and VOD with VOD having slightly higher bit rates.

Table A -- 9. ITU-T G.1080 Examples of Minimum Bit Rate Objectives for H.264 video.

SDTV Broadcast Video	1.75 Mbps CBR
SDTV VOD	2.1 Mbps CBR
HDTV Broadcast Video	10 Mbps CBR

There are additional audio bit rate objectives. Audio-video synchronization objectives are also given: 15 milliseconds maximum audio lead and 45 milliseconds maximum audio lag. [ATSC Doc. IS-191, "ATSC Implementation Subcommittee Finding: Relative Timing of Sound and Vision for Broadcast Operations Advanced Television," 26 June 2003.]

G.1080 describes network impairments, and quotes some packet loss requirements from Broadband Forum TR-126 in an appendix. Requirements for presented text quality are presented. Informative descriptions are given of QoE for control, including channel change time and VOD trick-play, as well as for browser and navigation.

F. ITU-T G.1010-2001

ITU-T G.1010 defines end-user multimedia QoS categories. This document defines broad QoS categories for many different IP-based services and doesn't focus on IPTV. The focus is on network performance. Key parameters affecting the user are stated in terms of delay, delay variation, and information loss from compression and packet loss. There is little in this document specific to IPTV.

G. ITU-T G.1050-2007

ITU-T G.1050 [32][33] defines profiles A, B, and C for IP networks, and lists performance thresholds as shown in the **Error! Reference source not found.**Table A -- 7, A—8, and **Error! Reference source not found.**A -- 9 below. G.1050 also presents a range of impairment conditions, discusses aggregating network performance across multiple links, and stratifies performance by the likelihood of occurrence of network impairments.

Table A -- 10. ITU-T G.1050 Table 2 – Impairment ranges for well-managed network (profile A)

Impairment type	Units	Range (min to max)
One-way latency	ms	20 to 100 (regional) 90 to 300 (intercontinental)
Jitter (peak-to-peak)	ms	0 to 50
Sequential packet loss	ms	Random loss only (except when link failure occurs)
Rate of sequential loss	sec^{-1}	Random loss only (except when link failure occurs)
Random packet loss	%	0 to 0.05
Reordered packets	%	0 to 0.001

Table A --11. ITU-T G.1050 Table 3 – Impairment ranges for partially-managed network (profile B)

Impairment type	Units	Range (min to max)
One-way latency	ms	20 to 100 (regional) 90 to 400 (intercontinental)
Jitter (peak-to-peak)	ms	0 to 150
Sequential packet loss	ms	40 to 200
Rate of sequential loss	sec^{-1}	$\leq 10^{-3}$ (Note)
Random packet loss	%	0 to 2
Reordered packets	%	0 to 0.01
NOTE – Sequential packet loss occurs once every 1000 seconds.		

Table A -- 12. ITU-T G.1050 Table 4 – Impairment ranges for unmanaged network (profile C) (Note 1)

Impairment type	Units	Range (min to max)
One-way latency	ms	20 to 500
Jitter (peak-to-peak)	ms	0 to 500
Sequential packet loss	ms	40 to 10'000
Rate of sequential loss	sec^{-1}	$\leq 10^{-1}$ (Note 2)
Random packet loss	%	0 to 20
Reordered packets	%	0 to 0.1
NOTE 1 – This table represents levels for a normally operating unmanaged network. Impairment levels for impairment condition H may exceed the ranges in this table to account for disaster conditions. NOTE 2 – Sequential packet loss occurs 1 every 10 seconds.		

APPENDIX B -- Voice over IP (VoIP) QoS and QoE

A summary of desirable network quality indicators for VoIP is in Table B -- **13**. VoIP is fairly tolerant of network errors, but requires bounded delay for conversations. The throughput required for an individual VoIP stream is low and so is not listed here.

Table B -- 13. Desired Network Performance Thresholds for VoIP service. NOTE: These numbers are end-to-end performance at the input to the encoder after all error correction including forward error correction and retransmission.

VoIP Test Parameter	**Desired value**
MOS-A	$\geq$ 3.6
Latency	$\leq$ 150 millisecond
Jitter	$\leq$ 50 millisecond
Packet loss rate (PLR), after all error correction	$\leq$ 1 percent

For VoIP, it is recommended to use the E-model R-factor [11], or the PESQ algorithm [12][13] to estimate audio quality. For VoIP, a MOS of 4 is considered PSTN quality, 3 is reasonably acceptable and 2 or less is not tolerable.

The quality of voice quality transmitted over the public-switched telephone network (PSTN) is often assessed with the ITU-T G.107 E-model [11]**.** The E-model derives voice quality in terms of analog parameters including signal power, noise power, echo, delay, equipment impairment. The E-model expresses quality by the parameter R, which ranges from about 0 to 100, with higher scores better. In summary,

$$R = Ro - Is - Id - Ie + A$$

where *Ro* represents the basic signal-to-noise ratio, *Is* represents impairments occurring simultaneously with the voice signal, *Id* represents impairments caused by delay, and *Ie* represents the impairments caused by low bit rate codecs. *A* is the advantage factor and can compensate for other advantages to the user.

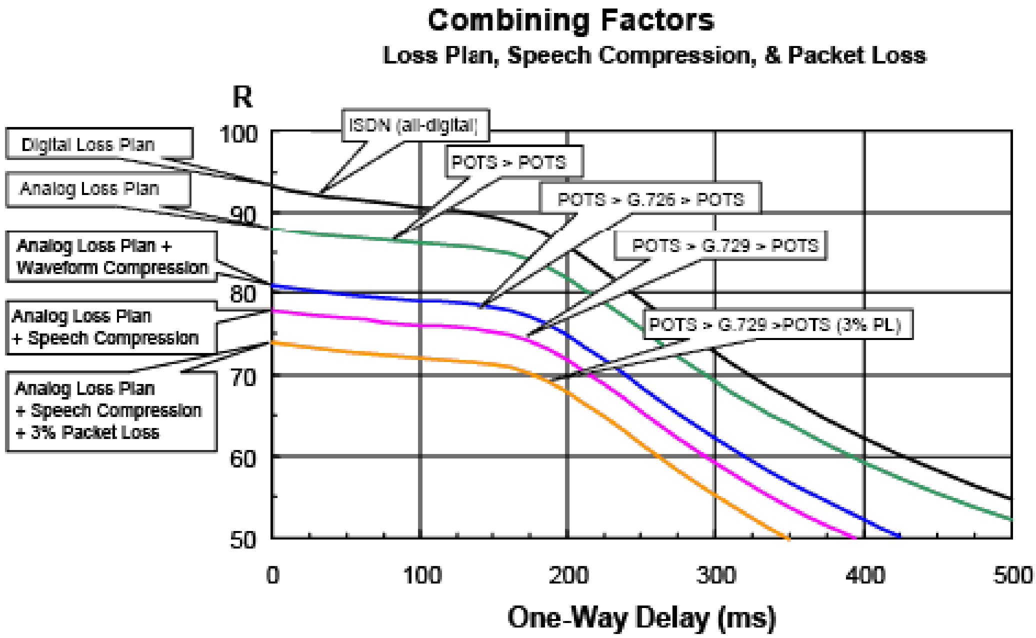


Figure 1. DSL Forum TR-126, Figure 20, R factor vs. delay for multiple distortion factors: loss plan (loudness), compression coding, packet loss, and delay.

Voice quality is often expressed by subjective rating of mean opinion score (MOS), with MOS ranging from 1 (poor) to 5 (perfect). A MOS of 4 is considered PSTN quality, 3 is reasonably acceptable and 2 or less is not tolerable. Subjective Voice Quality is often estimated using the full-reference calculations specified in ITU-T Recommendation P.862, Perceptual Evaluation of Speech Quality (PESQ) [12][13]. VoIP quality can be expressed as listening quality only (MOS-LQ), or conversational quality (MOS-CQ) which includes the effects of delay

ITU-T G.1050 [32][33] shows how a range of network conditions impacts VoIP quality.

APPENDIX C: IP Video Metrics

A. IP Video Performance Thresholds

This subsection discusses performance thresholds or targets, overviews standards with IP video performance thresholds, and presents some example numbers for IP video performance thresholds. These numbers are approximate, and are presented here as a starting point. Following this is a brief review of performance targets in a few related standards.

An example of the performance thresholds that may be needed in the IP video *network* is shown in Table C -- 14. These are example numbers that can vary with different network conditions and applications, as is discussed further below the table. Note that performance thresholds may also need to be specified for different locations and layers of the IP video service; *i.e.*, the ITF buffering, or the channel change (zap) time.

Table C -- 14. Example of possible performance thresholds for broadcast – quality IP video service. NOTE: These numbers are end-to-end performance at the input to the encoder after all error correction including forward error correction and retransmission.

IPTV Metric	Performance Threshold
Video Bit-Rate (CBR throughput for streaming video)	100 kbps to 15 Mbps. Defined separately for HDTV, SDTV, multimedia, mobile.
Packet loss rate (PLR), after all error correction	$\leq 10^{-3}$ to 10^{-6}
Jitter	$\leq$ 100 millisecond
MOS-A, MOS-V, MOS-AV	$\geq$ 3.6
Channel Change Latency	300 milliseconds to 2 seconds

The thresholds in Table C -- 14 could be defined differently for different TV service levels: HDTV, Broadcast-quality SDTV, Multimedia (PC-viewed), and Mobile. The metrics in Table C **-- 14** are discussed further in the following.

Video Bit-Rate. The minimum video bit rate may be specified so that the video is not overly compressed. This is done, for example, in ITU-T G.1080 [5]. However, it should be recognized that this is a simplistic way of specifying compression performance. Many factors impact compression: content, encoder settings, etc., and the output bit rate may be VBR or CBR.

Packet loss rate (PLR), after all error correction. Existing standards specify differing packet loss rates for IP video. ITU-T Y.1541 recommends a PLR threshold of 4×10^{-7} for access distribution at 3 Mbps assuming that 10 performance hits per day are tolerated. Broadband Forum TR-126 presents a range of PLR thresholds; for H.264 compression PLR is thresholds are 6×10^{-6} (SDTV) to 1.2×10^{-6} (HDTV). For digital television, ITU-T J.241 recommends that PLR $\leq 10^{-5}$ for excellent service quality (ESQ) and PLR $< 2 \times 10^{-4}$ for intermediate service quality (ISQ). Overall, existing standards recommend PLR thresholds from 2×10^{-4} to 4×10^{-7}, which are rounded to the order of magnitudes presented in Table C -- 14.

Many loss events may be essentially unnoticeable, depending on the error resilience of the encoder and decoder. Loss that impacts only a still part of the picture can be masked by a decoder treating these as skipped macroblocks and then simply displaying the same part of the picture that was in previous frames. Survey data has shown that many users don't notice loss events [42]. Because of this, the range of PLR in Table C -- 14 is rounded-up somewhat from the range calculated in the standards.

Jitter, or packet delay variation, can be induced by encoding, transmit and receive buffering, and network transmission. Jitter may be measured at the IP-layer and the MPEG stream layer. The level of tolerable jitter is largely determined by the size of the buffer used in the user terminal device. Large buffers can tolerate high jitter levels, but they can also cause high “channel change” delays for switching between content streams.

Broadband Forum TR-126 recommends a jitter threshold of 50 milliseconds, but current IP video set-top boxes can reportedly tolerate up to 50 to 150 milliseconds jitter. CPE for services on a groomed network, such as IPTV, may only tolerate low jitter levels. CPE for services on the open Internet, such as VoIP, may be able to tolerate high jitter levels. ITU-T G.1050 recommends a jitter threshold of 50 milliseconds for well managed networks and 150 milliseconds for partially-managed networks.

MOS-A, MOS-V, MOS-AV Mean Opinion Scores (MOS) for audio (-A), video (-V), and for audio/video (-AV) can be determined subjectively or algorithmically estimated objectively. There are many factors that can cause MOS scores and the thresholds associated with them to vary. Different TV levels have different user expectations and MOS for these are scored differently: HDTV, Broadcast-quality SDTV, Multimedia (PC-viewed), and Mobile. For example, the same decoded picture quality generally will get a higher subjective MOS score if shown on a mobile than on a big screen, because the expectations are lower and the screen is smaller on the mobile. On the other hand, HDTV service may only be acceptable with high MOS scores. Other factors affect MOS: source quality, content type, scene complexity, motion, viewing circumstances, etc. MOS thresholds may also need to vary with subscription level and pricing.

Channel Change Latency only applies to linear broadcast services and may be measured at the display or in the network control. If network control signals are measured, then channel change is the time between the initial IGMP signaling is sent to initiate the channel change and the time that the video stream begins to be received. A threshold for channel change latency until the signal is displayed may be up to several seconds; while a threshold for channel change latency for network control may be about 350 milliseconds.

B. Stream Statistics

Video is often carried in MPEG transport streams (TS). MPEG TS contain time stamps, sequence numbers, and program associations for packetized video streams. ETSI TR 101-290, SCTE 142 2007, and ATSC A/78A define stream errors; these include sync errors (PCR), continuity count (CC) errors, sub-program (PID) not present, and table errors (PAT and PSI errors).

C. Quality Layers

The ATIS IPTV Interoperability Forum (IIF), QoS Metrics (QoSM) Committee has been standardizing QoS and QoE metrics for IPTV. IIF QoSM has a conceptual IPTV quality stack with four layers: Transmission, Media Stream, Content, and Transaction quality. Transmission and stream quality can be monitored throughout the network. Transaction quality is a big part of the IPTV Quality of Experience (QoE) and includes channel change "zap" time, delay from request to delivery of a service such as VOD, and even the ease of navigation through a program guide or use of a service menu. Servers can deliver utilization statistics and counts of incomplete or delayed requests. Servers that deliver necessary related services should be monitored; such as digital rights management

(DRM), and subscription management servers. Sessions can be tested by connecting a test probe to the service, mimicking user interactions such as channel changes, and tracking performance statistics.

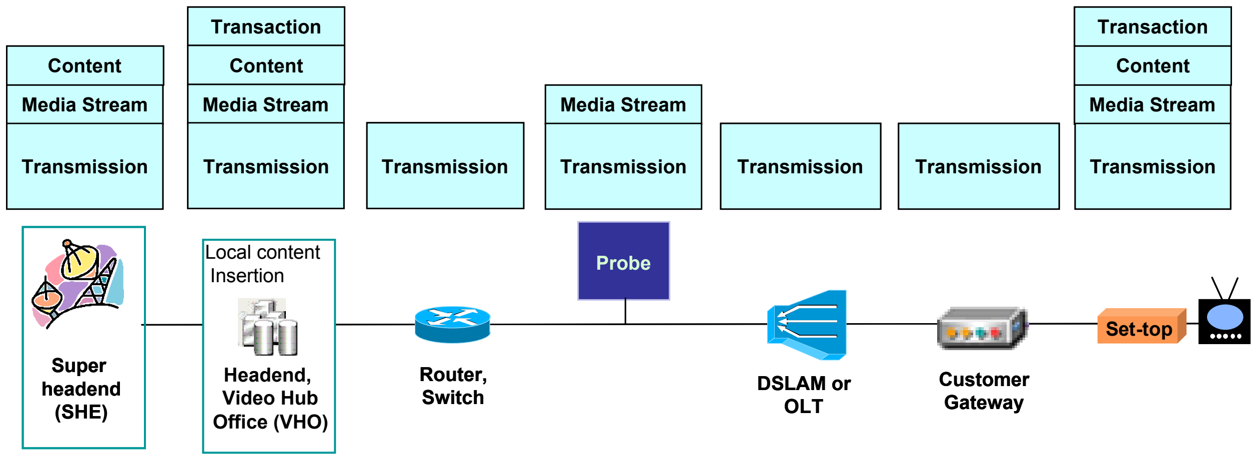


Figure C -- 2. Quality layers and measurement points.

Table C -- 15. Examples of some QoS metrics and QoE indicators at different quality layers.

Lay	**QoS**	**QoE**
Transaction	IGMP leave/join time	Channel change (zap) time
	EPG functionality	Ease of use
	Set-top boot time	Customer experience
	DRM errors	Service functioning
	Server overload	
Content	Video bit rate	Picture quality
	Audio-video synch	Audio quality
	Coding parameters	Multimedia quality
Media stream	Timing, timestamp, errors	Video artifacts
	Set-top buffer overflow/underrun	
	Loss of mapping tables	
Transmission	Packet loss	Network alarms
	Delay	Availability
	Jitter	

D. IP Video Quality of Experience (QoE) Metrics

QoS metrics are somewhat low-level and generally report machine-measurable network performance. At a higher-level, a Quality of Experience (QoE) metric is, ideally, a measure that shows exactly how a subscriber would rate the service quality. QoE may be estimated algorithmically with a machine, or by users themselves with customer satisfaction surveys. Some QoE measures such as video picture quality are very complex and account for human physical and psychological patterns; for example, humans focus on certain parts of a picture.

An example listing of QoE metrics is:

- The following may be measured objectively:
  - Video quality
  - Audio quality
  - Audio-Video synchronization (lip synch)
  - Multimedia quality
  - Overlay/supplemental application quality
  - Quality of text subtitling and captioning [ITU-T F.700]
  - Synchronization of subtitling and video
  - Loudness variation
  - Transactional delay, such as channel change delay, and VOD trick play
  - EPG response times
- The following may be measured with a survey:
  - Bill presentation quality
  - Customer relations
  - Ease of use of service
  - Ease of use of device
  - Ease of use of EPG
  - Perceived service value

- Service Performance/Key Quality Indicators (KQI)
    - Rate of occurrence and severity of high-delay transactions
    - Rate of occurrence and severity of failed transactions
    - Rate of occurrence and severity of Degraded Service Quality (DSQ) events
    - Rate of occurrence and duration of service outage, or availability
    - Rate of occurrence and duration of program off-air

DSQ - Degraded Service Quality event, is a noticeable impairment of the audio quality, video quality or service response time. [TM Forum GB938]

APPENDIX D: Table of Acronyms and Abbreviations

Acronym	Meaning
AAA	Authentication, Accounting, and Authorization
ADI	Asset Distribution Interface
API	Application Programming Interface
APOD	ATIS Point of Deployment Module
ATIS	Alliance for Telecommunications Industry Solutions
ATM	Asynchronous Transport Mode
BPI	Broadband Performance Index
CBR	Constant bit rate
CDDC	Consumer Domain Device Configuration
CDR	Committed Data Rate
CIR	Committed Information Rate
CM	Component Missing
CO	Central Office
COD	Content on Demand
CPE	Customer Premises Equipment
DLNA	Digital Living Network Alliance
DNG	Digital Network Gateway (DSL Modem or ONT)
DPI	Digital Program Insertion
DR	Draft Revision of a standard
DRM	Digital Rights Management (Copyright)
DS	Draft Standard
DSL	Digital Subscriber Line, any variant
DSQ	Degraded Service Quality
DVB	Digital Video Broadcast, European Standards
EAP	Extensible Authentication Protocol
EAS	Emergency Alert Service
ECM	Entitlement Control Messages
EIR	Excess Information Rate
EMM	Entitlement Management Messages
EPG	Electronic Program Guide
ES	Errored Second
ETSI	European Telecommunications Standards Institute
FCC	Federal Communications Commission
FEC	Forward Error Correction
FLUTE	FiLe delivery over Unidirectional Transport
FTTN	Fiber To The Node
FTTP	Fiber To The Premises
Gbps	Giga-bits per second
GigE	Gigabit Ethernet
GMI	Global MultiService Interoperability
GR	Generic Requirements
IDSA	IIF Default Scrambling Algorithm
IGMP	Internet Group Management Protocol
IMS	IP Multimedia Subsystem
IO	Intermediate Office

Acronym	Meaning
IPG	Interactive Program Guide
IPTV	TV using Internet Protocol
IPPM	IETF IP Performance Metrics group
ISS/A	IPTV Security Solution/Authentication
ISSI	IPTV Separable Security Incubator
ITF	IPTV Terminal Function (Set-top box)
ITU	International Telecommunication Union Telecommunication Standardization
kbps	kilo bits per second
kft	kilofeet (thousands of feet)
KQI	Key Quality Indicator
LB	Letter Ballot
LSP	Label Switched Path, in MPLS
Mbps	Mega-bit per second
MHP	Multimedia Home Platform
MLT	Metallic Loop Test
MOS	Mean Opinion Score
MPEG	Motion Pictures Experts Group
MPEG TS	MPEG2 Transport Stream
MPLS	Multi-Protocol Label Switching
MSO	Multiple System Operator
MSF	MultiService Forum
NOI	Notice of Inquiry
NPVR	Network Personal Video Recorder
NHTSA	National Highway Traffic Safety Administration
OLT	Optical transceiver at CO
ONT	Optical transceiver at customer location
ONU	Optical Network Unit
PEG	Public, Education, and Government Channels
PESQ	Perceptual Evaluation of Speech Quality
PID	MPEG TS Program Identification
PIP	Picture in Picture
PKI	Public Key Infrastructure
PLR	Packet loss rate
PMT	MPEG2 TS Program Map Table
POA	Program Off Air
PON	Passive Optical Networks
POTS	Plain Old Telephone Service
PPP	Point-to-Point Protocol
PPV	Pay Per View
PRQC	ATIS Network Performance, Reliability, and QoS Council
PSD	Power Spectral Density
PVR	Personal Video Recorder (TiVo)
PQM	Perceptual Quality Metric
QoE	Quality of Experience
QoS	Quality of Service
RADIUS	Remote Authentication Dial-In User Service
RTCP	Real-Time Control Protocol

Acronym	Meaning
RTP	Real-Time Protocol
RTSP	Real-Time Streaming Protocol
SDP	Session Description Protocol
SEE	Secure Execution Environment
SHE	Super Head-End
SI	Service Information
SIP	Session Initiation Protocol
SP	Service Provider
SSE	Separable Security Element
TBD	To Be Determined
TCP	Transmission Control Protocol
TNC	Technically Non Conformant
TRQ	Technical Requirements
TS	Transport Stream
TSK	Telcordia Standards Knowledgebase
UDP	User-Datagram Protocol
VBR	Variable bit rate
VDSL	Very High Speed DSL
VHO	Video Hub Office
VLAN	Virtual local-area network
VOD	Video on demand
VoIP	Voice on Internet Protocol
VPLS	Virtual Private Line Service (VLAN on MPLS)
VPWS	Virtual Private Wire Service
VSO	Video Serving Office
WDM	Wavelength Division Multiplexing
WM	Windows Media
WT	Working Text
XML	eXtensible Markup Language

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